



## **User's Manual**

## **Internet IP Telephone**

**Model No.: SP5100/S**

Website: <http://www.micronet.info>

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# **1. Overview of the IP Phone**

IP-Phone is a full-featured IP-based telephone set via Ethernet base communication. Over the office LAN connection, it provides IP-PBX solution such as station-to-station call, IP call and local PSTN/PBX Extension call via PSTN Gateway.

IP-Phone provides two 10/100BaseT switch/hub RJ-45 ports allow connecting to office LAN and PC simultaneously. It is compatible with ADSL or Cable Modem provided by ITSP, ISP or Carrier Company to provide VoIP services to residential and SOHO application.

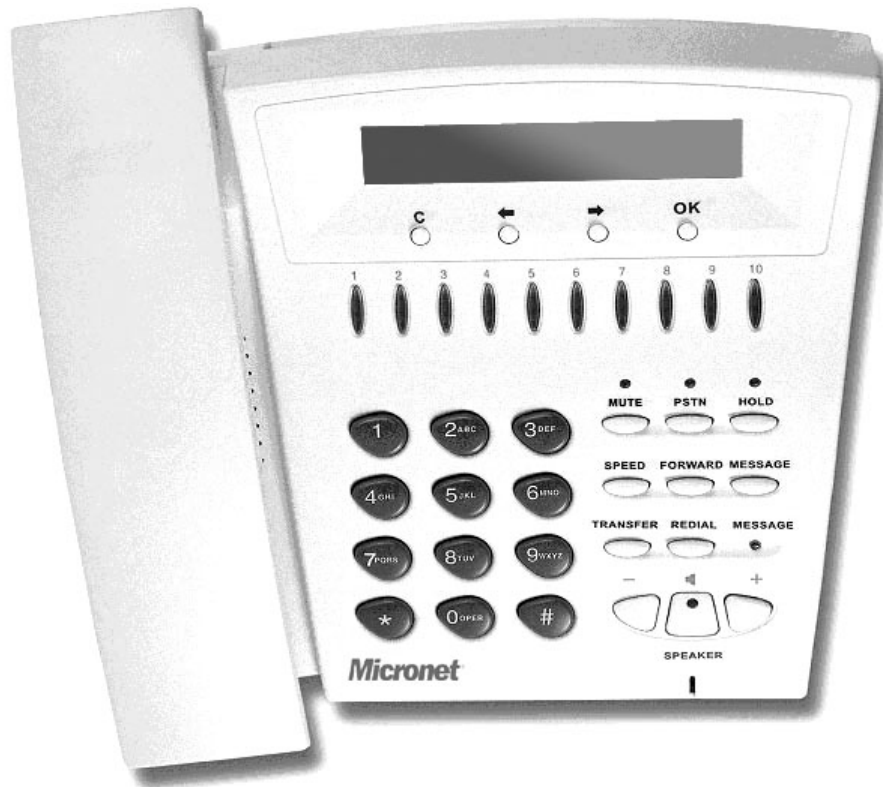
IP-Phone is also an integrated Analog Phone provides IP call or PSTN call selection. When external power is down, it can be a Plain Old Telephone set (POTs).

It provides internal high-quality speakerphone, programmable keys and feature buttons.

IP-Phone also embedded with a dot matrix of two lines 24 characters LCD, which can display date and time, calling party name, calling party number, and digits dialed and etc.

## 1.1. Hardware Overview

### Front View and Keypad function



- **LCD:** 2 lines, 24 character Dot Matrix display.
- **C:** Jump out current LCD menu.
- **←:** Move to previous selection or clear previous data.
- **→:** Move to right or next selection.
- **OK:** Press OK to confirm the modification.
- **Direct Line (DL) Button 1 – 10:** User press DL button after off-hook to do speed dial according to phone book data from 1-10 (please refer to LCD configuration-3.Phone Book; Advanced Configurations via Telnet- 10.[pbook] command, or Web Configuration-Phone Book chapter.
- **Number 1 –10, \* and #:** The function is the same as the general phone set.

### Corresponding list of keypad and symbol:

#### (1) Character mode:

1	"1"
2	"A" ; " B" ; "C" ; "2"

3	"D" ; "E" ; "F" ; "3"
4	"G" ; "H" ; "I" ; "4"
5	"J" ; "K" ; "L" ; "5"
6	"M" ; "N" ; "O" ; "6"
7	"P" ; "Q" ; "R" ; "S" ; "7"
8	"T" ; "U" ; "V" ; "8"
9	"W" ; "X" ; "Y" ; "Z" ; "9"
*	" " ; "@" ; "_" ; "-" ; "." ; "!" ; "?" ; "*" ; "+" ; "\$" ; "%"
0	"Space" ; "0"
#	"#"

## (2) Digit mode:

1	"1"
2	"2"
3	"3"
4	"4"
5	"5"
6	"6"
7	"7"
8	"8"
9	"9"
*	" "
0	"0"
#	

- **MUTE:** Mute the voice of MIC and let others can't hear from user in communication.
- **PSTN:** Press PSTN to switch IP-Phone as PSTN or IP Phone Mode. In PSTN mode, "PSTN" characters will be displayed on LCD left bottom side, then users can dial out as if standard telephone set in PSTN; in IP Phone mode, "PROXY" characters will be displayed on LCD left bottom side.

### Note:

1. When IP-Phone is in PSTN mode, only **PSTN** and **SPEAKER** function key can work.
2. On LCD will display "...Incoming Call..." to inform user when IP-Phone has both IP and PSTN side incoming calls.
3. If in communication with IP side, user can press HOLD to hold IP side, then press PSTN to pick up PSTN side, after that can press HOLD again to retrieve IP side.
4. If in communication with PSTN side, user must hang up PSTN side before pick up IP

side.

- **HOLD:** To hold a call, after press HOLD button, both sides will hear hold tone.
- **SPEED:**
  1. Press SPEED and number (Phone book index) after off-hook can do speed dial according to phone book data (please refer to **3.LCD configuration-6. Phone Book** or Advanced Configurations via 4.Telnet- **10.[pbook]** command).
  2. Switch input mode between character mode or digit mode, e.g., when user wants to input phone number can press **SPEED** to switch input mode as digit mode; when user wants to input name can also press **SPEED** to switch input mode as character mode.
- **FORWARD:** Forward an incoming call to another IP device. (Please refer to LCD configuration-Forward Type) (\*\*)
- **MESSAGE and its indicated LED light:** When having missed incoming calls, the MESSAGE LED will be flashing. User can check the information of missed calls by pressing the MESSAGE button. (\*\*)
- **TRANSFER:**
  1. Transfer a call to the third site. When A and B are in communication, A wants to transfer this call to C, A can press **TRANSFER** button, now B will hear hold tone, and A will hear dial tone, then A can press phone number of C, after C picks up, A can talk with C, after A hangs up, B and C can be connected.

Note:

1. A cannot press phone number of C before hearing dial tone.
2. If A press TRANSFER and number, press TRANSFER again can cancel transferring and retrieve call with B.
3. If A press TRANSFER and number of C, C is ringing, press TRANSFER again C will return to standby mode and A can retrieve call with B.

2. Change characters to be capital or lowercase: when pressing **TRANSFER** before press letters can switch input type of letters.

- **REDIAL:** Redial the last outgoing call.
- **+ And -:** Adjust the voice volume heard of communication.
- **SPEAKER:** Hand free mode. User can talk without picking up handset.

**Note:**

All function keys mentioned above (except dialing keypad) are effective only in IP Phone mode.

**Back View**



- ❶ **DC 9V:** DC 9V power input outlet
- ❷ **LAN:** RJ-45 connector, connected directly to the Hub through the straight CAT-5 cable.
- ❸ **PC:** RJ-45 connector, connected directly to the PC through the straight CAT-5 cable
- ❹ **Line:** RJ-11 connector, connected directly to the PSTN analog line.

**Note:**

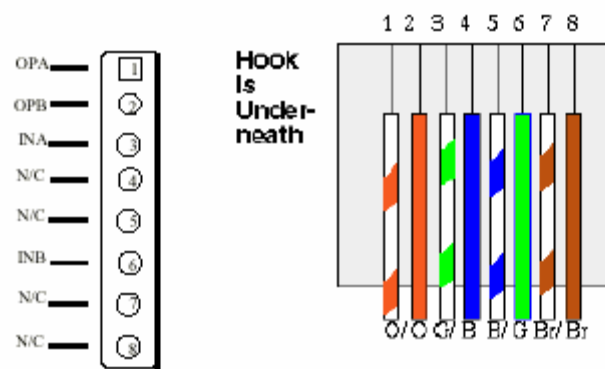
There are two LED indicated lights: LINK/ACT and 10/100 for LAN port and PC port. When network status is regular, LED of LINK/ACT will light on; when IP-Phone is transmitting or receiving data, LED will be flashing; when transmit rate is in 10 mbps or 100mbps, LED of 10/100 will light off or light on.

**Specification of connector**

Ethernet Port :

Ethernet port is for connecting IP-Phone to network, transmit rate supports 10/100 Base-T.

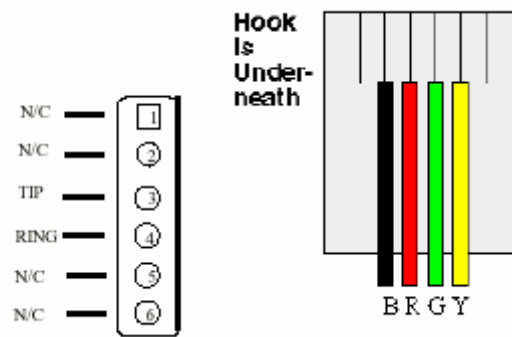




Ethernet connector ( LAN )

RJ-11connector :

RJ-11 connector is for connecting IP-Phone with PSTN.



RJ11connector

## 1.2. Software Features and Specification

### Application:

- ISP/ITSP (Internet Telephony Service Provider)
- IP-PBX with office telephony services
- Multi-nation enterprise communication
- SOHO Telephony

### Calling Features

- Call Hold
- Call Transfer
- Call Forward (\*\*)
- 10 sets last number Redial
- Ten configurable speed dials

### Network Supported

- Fixed IP
- Dynamic Host Configuration Protocol (DHCP)
- PPPoE connection (When PPPoE disconnect, IP-Phone can automatically re-connect)
- Behind NAT IP Sharing Device
- Support QOS by setting DSCP (Differentiated Service Code Point) parameters of VoIP packet
- Support 802.1p1q (VLAN) (\*\*)

### Audio Features

- G.711 a/μ-Law, G.723.1, G.729, G.729a
- VAD, CNG
- G.165/G.168 compliant echo cancellation
- Programmable Dynamic Jitter Buffer
- Bad Frame Interpolation
- Gain/Attenuation Settings

### **Provisioning and Configuration**

- SIP (RFC3261) compliance
- LCD configuration password protection
- Provide Proxy Mode or Peer-to-Peer Mode (Non Proxy Server needed) selection
- Ring tone, Speaker and Handset volume adjustable
- Dial path selection (PSTN or IP mode)
- Support DNS server inquiry

### **Management Features:**

- Software Upgrade: TFTP/FTP download
- Three easy ways for system configuration
  1. LCD Front Panel
  2. Web Browser
  3. TELNET

### **Certification**

CE, 3C

### **1.3. Physical specification & Environment**

**Dimension:**

215mm(W) x 71mm(H) x 198mm(D)

**Weight:**

834 grams (unit)

**Power Supply:**

DC 9V output adaptor, AC 120Vac input

**Operation Environment:**

Humidity: 10 to 90 % (Non-condensing)

Operational Temperature: 0 to +40 °C

**Storage Environment:**

Humidity: 10 to 90 % (Non-condensing)

Storage Temperature: -10 to +50 °C

**Analog Specification:**

2 wires (RJ-11) \*1 (Voltage: -48V)

AC Impedance 600Ω

## 2. Configuring the IP-Phone - LCD Menus

### Note:

1. After any configuration is made for the IP-Phone , user has to do Reboot in the selection 7 “Reboot”.
2. We suggest user to set IP address via LCD menu 5→2.3.4.5 first, then go to chapter 3 to do other detail configurations via web browser.

### 2.1.Initialize IP-Phone

When power on the IP-Phone, the LCD screen shows as below. Now IP-Phone is running Boot sector program.

IP-Phone Board Start Booting
---------------------------------

When IP-Phone finishes boot program initialization. User can see flashing greeting as below:

System Initializing.....
--------------------------

Then IP-Phone get into standby mode:

IP-Phone Proxy 10:10:10 AM
-------------------------------

The main LCD screen would be shown as similar as above. “Proxy” means the IP-Phone is in Proxy Mode, and when IP-Phone is connected to SNTP server, on LCD will show current time captured from SNTP server.

When IP-Phone is under peer-to-peer mode, on LCD will show “P2P” instead of “Proxy”.

IP-Phone P2P 10:10:10 AM
-----------------------------

After pressing the PSTN button, the “Proxy” or “P2P” will be replaced by “PSTN”. Please

notice that user must plug PSTN line in RJ-11 port when IP-Phone is in PSTN mode. (\*\*)

IP-Phone
PSTN 10:10:10 AM

Press ← or → to enter configuration mode then press OK button to enter sub menus; press C can jump out current menu to previous level.

1. Call List (\*\*)
2. Forward Type
3. Phone Book
4. Ringer Settings
5. Network
6. Advanced Settings (protected by password)
7. Reboot

**Note:**

**(\*\*): May be not available right now, please contact with your distributor.**

## 2.2. LCD Menu Configuration

User can set the following configurations by LCD keypad.

Note:

1. Press **TRANSFER** before input data can switch characters to be capital or lowercase.
2. Press **SPEED** before input data can switch input mode as character mode or digit mode; for example, user wants to enter IP address, after pressing **SPEED** can enter digits directly.
3. When user is inputting data, press **C** will jump out to menu list, press **←** will clear previous input data.

### 1. Call List (\*\*)

If there is an unanswered IP call, it will be kept in message box. MESSAGE LED will be flashing until user press MESSAGE to check miss call and re-press MESSAGE to return to main screen.

Missed Calls : to see all missed calls in message box.

Received Calls : to see all received calls in message box.

Dialed calls: to see all dialed calls in message box.

### 2. Forward Type (\*\*)

There are 3 selections in Forward type, user must select under which condition to forward calls.

#### 1. Busy

When IP-Phone is in busy status, the incoming call will be forwarded to the assigned phone number.

##### A. Activate

Enter a forwarded phone number to activate busy forward function.

##### B. Deactivate

Deactivate Busy Forward function.

#### 2. No Response

When IP-Phone hasn't been picked up for around 10 seconds, the incoming call will be forwarded to the assigned phone number.

##### A. Activate

Enter a forwarded phone number to activate no response forward function.

**B. Deactivate**

Deactivate No Response Forward function.

**3. Unconditional**

It is included the above two types. Whether the IP-Phone is in which status, calls will be automatically forwarded to the assigned phone number.

**A. Activate**

Enter a forwarded phone number to activate Unconditional Forward function.

**B. Deactivate**

Deactivate Unconditional Forward function.

**3. Phone Book**

**1. List**

List all records of name, telephone number, and IP address in the phone address book.

**2. Edit/Delete**

Edit or delete a record of name, telephone number, and IP address of the phone address book.

**3. New**

Add a new record of name, telephone number, and IP address of the phone address book.

**4. Ringer Settings**

**1. Volume**

User can adjust ring volume by press ◀ or ▶ on the keypad to decrease or increase ringer volume.

**2. Style**

There are three tone styles for IP-Phone. Move the ">" symbol by press ◀ or ▶ on the keypad to select the tone style preferred, then press OK to confirm it.



## 5. Network

### 1. Information

User can press ⬅ or ➡ to check current network status:

- Mode: Display current network connection mode of IP-Phone to be Static (Fixed IP), DHCP, or PPPoE.

Note:

When IP-Phone is under DHCP mode, then change to Static mode, the following items: IP address, Subnet Mask, Default Gateway, will display empty, after reboot, user can see information again.

- IP, Mask, Gateway: display current IP information.

### 2. Network Mode

Set network mode of IP-Phone to be Static (Fixed IP), DHCP, or PPPoE.

### 3. IP address

Set IP address of IP-Phone.

### 4. Subnet Mask:

Set subnet mask address of IP-Phone.

### 5. Default Gateway

Set default gateway address of IP-Phone.

### 6. Domain Name Server

Set IP address of Domain Name Server. Once IP-Phone can connect to DNS server, user can set URL address for Proxy server or Phone book instead of IP address.

### 7. PPPoE Configuration

- User Name  
Set PPPoE connection authentication user name.
- Password  
Set PPPoE connection authentication password.
- Auto Re-connect  
Choose ON or OFF to enable or disable this function. If user enables this function, after PPPoE disconnected, IP-Phone will automatically reboot to re-connect, and after reboot, if IP-Phone still can't connect with server,

IP-Phone will keep trying to connect. On the other hand, if user disables this function, IP-Phone won't reboot and keep trying to connect.

## 8. SNTP Configuration

### (1) SNTP Mode:

User can set SNTP function to be on or off, which means IP-Phone will capture current time from SNTP server or not.

### (2) SNTP Server:

User can specify a SNTP server for IP-Phone to capture current time.

### (3) Time Zone:

User can set time zone via pressing ◀ or ▶ according to the location IP-Phone is. For example, in Taiwan the time zone should be set as GMT+8:00.

## 9. Behind IP-Sharing

- If IP-Phone is behind IP sharing or NAT device, on IP sharing must enable "DMZ" function or set "Virtual Server" to open ports (UDP port: 5060 and 16384).
- User must enter public IP address of IP sharing.

## 6. Advanced Settings (protected by password)

Please Enter Password: (Default: Null)

### Note:

Default Password is empty, user can enter this sub-menu directly, once password has been set in 3.LCD Menu password, user must input password before enter this sub-menu.

If user forget password, please contact with your distributor, we will generate a specific password according to your MAC address of IP-Phone.

User can also try to configure IP-Phone via Telnet or Web browser with default IP address: 10.1.1.3. (This only works when default IP address hasn't been changed.)

## 1. SIP Settings

### 1. Connect Mode

Select SIP connection mode to be peer-to-peer mode or Proxy mode.

## 2. Proxy

- Proxy  
Set Proxy IP address or Domain Name.
- Outbound proxy  
Set Outbound Proxy IP address or Domain Name.
- Proxy port  
Set Proxy port for IP-Phone to send messages.
- Expire (in seconds)  
Set expire time of registration, in the duration of 2/3 expire time, IP-Phone will re-register to Proxy Server again.

## 3. User Info

- User Name  
Set User Name of IP-Phone to register to Proxy Server. If Proxy server doesn't request specific User name, please enter Line number here.
- Line Number  
Set Line Number of IP-Phone to register to Proxy Server.
- Password  
Set User Password of IP-Phone to register to Proxy Server. This configuration is not necessary, if Proxy server doesn't request client to set password, user only has to set User Name the same as Line Number.

## 2. Firmware Update

### 1. Download method

There are two methods to download new version file, please move the ">" symbol by press ⬅ or ➡ on the keypad to select TFTP or FTP method, then press OK to confirm it.

### 2. FTP/TFTP Sever

User has to offer one TFTP/FTP server IP Address and set this IP Address via keypad. The IP Address is necessary for upgrading IP-Phone new application rom file.

### 3. FTP Account

User has to input user name for FTP server login .It is necessary for upgrading IP-Phone new application rom file via FTP method.

#### 4. FTP Password

User has to input password for FTP server login .It is necessary for upgrading IP-Phone new application rom file via FTP method.

#### 5. File Name

User has to press the file name of new application rom file prepared for upgrading

#### 6. Application Version

Show current version of application.

#### 7. Start to Upgrade

Select YES or NO to start upgrade.

#### **Note:**

Download via LCD command can only upgrade new **application rom** file.

#### 3. LCD Menu Password

Set entry password of phone LCD menu.

### **7. Reboot**

Reboot machine. It is necessary and important for user to reboot IP-Phone after any configurations has been made. IP-Phone will ask user again before reboot.

### 3. Configuring the IP-Phone through Web Pages

The HTTPD web management interface provides user an easier way to configure rather than command line method through TELNET.

The configuration function and steps are similar with the way through command line. Please refer to the chapter 4- Configuring the IP-Phone through Telnet command lines for more detail information. Below is a guide for user to configure via web interface.

#### Step 1. Browse the IP Address predefined via Keypad

Please enter IP address (user have to set via LCD menu first) of IP-Phone in web browser. If user failed to set IP address via LCD menu, the default IP address of IP-Phone is 10.1.1.3, user can try to connect to IP-Phone via this default IP via web interface.



#### Step 2. Input the login name and password

Login name: root / administrator

Password (The same with TELNET): Null (just press confirm, no need to key in password in default value)



輸入網路密碼

請輸入您的使用者名稱與密碼。

網站: 192.168.2.103

範圍: IP-PHONE Web Configuration

使用者名稱(U): root

密碼(P):

☒ 將這個密碼存到您的密碼清單(S)

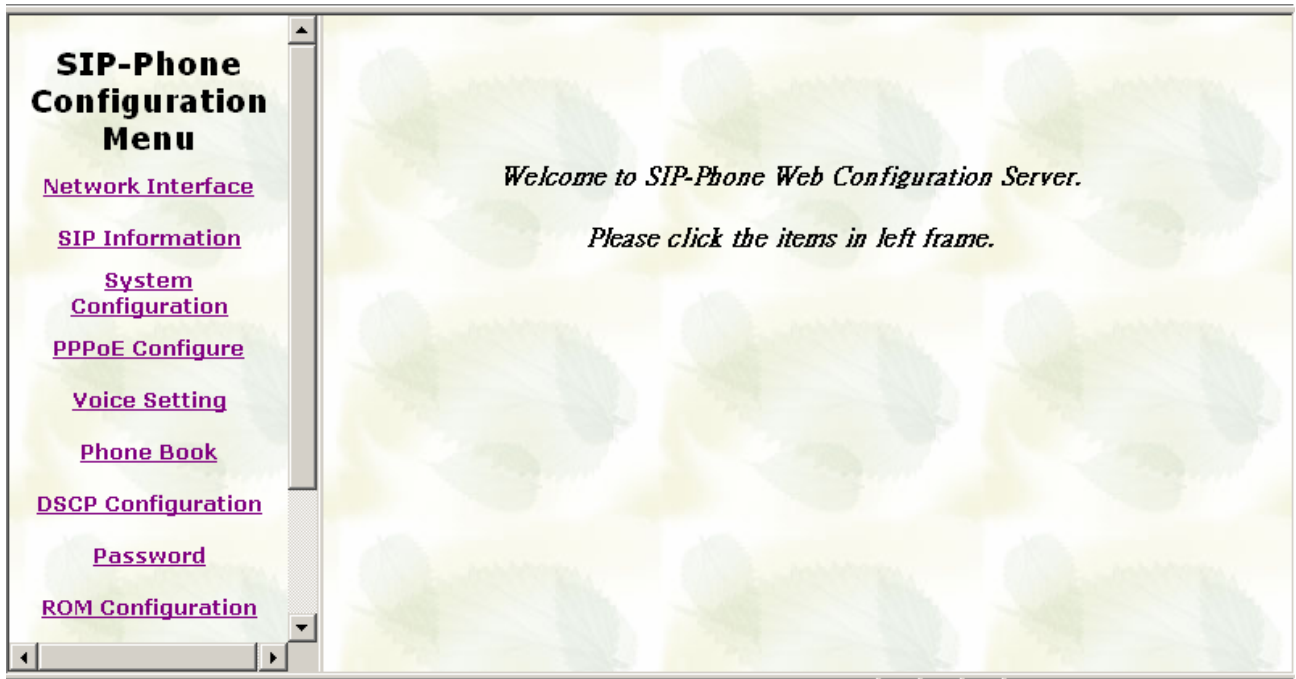
確定 取消

---

**Note:** User can set password later in 8.PASSWORD via web interface.

### Step 3. Enter the web interface main screen

After enter login name and password, user can see web interface main screen as below.



#### Step 4. Start to configure

Most important items are Network Interface, SIP Information, and Phone Book (in Peer-to-Peer mode). Please remember to configure these commands before start to work with IP-Phone.

#### Note:

After change any settings, please remember to **reboot** (in Reboot System) IP-Phone so that changes can take effect.

### 3.1. Network Interface

Please refer to chapter 4.7 [ifaddr] command.

The screenshot displays the 'SIP-Phone Configuration Menu' on the left sidebar and the 'Network Interface' configuration page on the right. The sidebar menu includes links for Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password, ROM Configuration, Flash Clean, Commit Data, and Reboot System. The 'Network Interface' page contains the following fields:

Network Interface	
IP Address:	192 . 168 . 4 . 103
Subnet Mask:	255 . 255 . 0 . 0
Default routing gateway:	192 . 168 . 1 . 254
Get IP Mode:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP <input type="radio"/> PPPoE
SNTP:	<input checked="" type="radio"/> enable <input type="radio"/> disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	8
IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
UPnP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
IP Sharing Server Address:	0 . 0 . 0 . 0
Primary DNS Server:	168 . 95 . 1 . 1
Secondary DNS Server:	168 . 95 . 1 . 2
<input type="button" value="OK"/>	

IP Address: Set IP Address of IP-Phone

Subnet Mask: Set the Subnet Mask of IP-Phone

Default routing gateway: Set Default routing gateway of IP-Phone

Get IP Mode: User has to set IP-Phone to use which network mode.

Fixed IP: User has to assign a fixed IP to IP-Phone.

DHCP: When DHCP function enables, IP-Phone will automatically search DHCP server after reboot.

PPPoE: If IP-Phone is working with PPPoE connection, user have to set related parameters in "PPPoE Configure" page.

Note:

If User set "Get IP mode" as DHCP or PPPoE, IP address, Subnet Mask, and Default routing gateway will become 0.0.0.0 and not allow to be configured.



### SIP-Phone Configuration Menu

- [Network Interface](#)
- [SIP Information](#)
- [System Configuration](#)
- [PPPoE Configure](#)
- [Voice Setting](#)
- [Phone Book](#)
- [DSCP Configuration](#)
- [Password](#)
- [ROM Configuration](#)
- [Flash Clean](#)
- [Commit Data](#)
- [Reboot System](#)

Network Interface	
IP Address:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Subnet Mask:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Default routing gateway:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Get IP Mode:	<input type="radio"/> Fixed IP <input checked="" type="radio"/> DHCP <input type="radio"/> PPPoE
SNTP:	<input checked="" type="radio"/> enable <input type="radio"/> disable
SNTP Server Address:	<input type="text" value="168"/> . <input type="text" value="95"/> . <input type="text" value="195"/> . <input type="text" value="12"/>
GMT:	<input type="text" value="8"/>
IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
UPnP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
IP Sharing Server Address:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Primary DNS Server:	<input type="text" value="168"/> . <input type="text" value="95"/> . <input type="text" value="1"/> . <input type="text" value="1"/>
Secondary DNS Server:	<input type="text" value="168"/> . <input type="text" value="95"/> . <input type="text" value="1"/> . <input type="text" value="2"/>
<input type="button" value="OK"/>	

### SIP-Phone Configuration Menu

- [Network Interface](#)
- [SIP Information](#)
- [System Configuration](#)
- [PPPoE Configure](#)
- [Voice Setting](#)
- [Phone Book](#)
- [DSCP Configuration](#)
- [Password](#)
- [ROM Configuration](#)
- [Flash Clean](#)
- [Commit Data](#)
- [Reboot System](#)

Network Interface	
IP Address:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Subnet Mask:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Default routing gateway:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Get IP Mode:	<input type="radio"/> Fixed IP <input type="radio"/> DHCP <input checked="" type="radio"/> PPPoE
SNTP:	<input checked="" type="radio"/> enable <input type="radio"/> disable
SNTP Server Address:	<input type="text" value="168"/> . <input type="text" value="95"/> . <input type="text" value="195"/> . <input type="text" value="12"/>
GMT:	<input type="text" value="8"/>
IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
UPnP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
IP Sharing Server Address:	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Primary DNS Server:	<input type="text" value="168"/> . <input type="text" value="95"/> . <input type="text" value="1"/> . <input type="text" value="1"/>
Secondary DNS Server:	<input type="text" value="168"/> . <input type="text" value="95"/> . <input type="text" value="1"/> . <input type="text" value="2"/>
<input type="button" value="OK"/>	

SNTP: Enable / Disable the Simple Network Time Protocol function

SNTP Server Address: Set SNTP Server Address

When SNTP server is available, enable IP-Phone SNTP function to point to SNTP server IP address so that IP-Phone can get correct current time.

GMT: Set time zone for SNTP Server time

User can set different time zone according to the location of IP-Phone. For example, in Taiwan the time zone should be set as 8, which means GMT+8.

IP Sharing: Enable it if IP-Phone is behind IP Sharing router.

UPnP: Enable it if IP sharing or NAT device supports UPnP function so that no need to configure IP sharing or IP-Phone when IP-Phone is behind NAT device.

IP Sharing Server Address: Set Public IP Address of IP Sharing router for IP-Phone to work behind IP sharing.

Primary DNS Server: Set Primary Domain Name Server IP address.

User can set Domain Name Server IP address. Once IP-Phone can connect with DNS server, user can specify URL address instead of IP address for Proxy and phone book IP address.

Secondary DNS Server: Set Secondary Domain Name Server IP address.

## 3.2. SIP Information

Please refer to chapter 4.13 [SIP] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'SIP Configuration' form on the right. The menu includes links for Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password, ROM Configuration, Flash Clean, Commit Data, and Reboot System. The SIP Configuration form contains the following fields:

SIP Configuration	
Run Mode:	<input type="radio"/> Peer-2-Peer <input checked="" type="radio"/> Proxy
Proxy IP Address:	192.168.4.192
Outbound Proxy:	null
Proxy port:	5060
Phone Book Search:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Prefix String:	null
Line Number:	1603
Line Account:	1603
Line Password:	1603
SIP port:	5060
RTP Port:	16384
Expire:	3600
OK	

Run Mode: Select IP-Phone to work under Peer-to-Peer mode or Proxy mode.

Proxy IP Address: Set Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to Network Interface).

Outbound Proxy: Set IP Address or URL address (Domain Name Server must be configured. Please refer to Network Configure) of outbound Proxy server.

Proxy port: Set Proxy port for IP-Phone to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.

Phone Book Search: enable/disable phone book search function. If user enabled this function, IP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, IP-Phone will make call to related IP address.

Prefix String: set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, IP-Phone will sent out messages as Account Name @"IP address of Proxy", if user set prefix as abc, IP-Phone will set out as abc123@"IP address of Proxy". This function is for special proxy server.

Line Number: identify one number for the IP-Phone to register to the Proxy.

Line Account: set user name of IP-Phone for registering. User can set user name and password for registering. If password is no need, please set user name the same as line number or IP-Phone won't register successfully.

Line Password: set password for registering.

SIP Port: set SIP UDP port.

RTP Port: set RTP port for sending voice data.

Expire: set expire time of registration. IP-Phone will keep re-registering to proxy server before expire timed out

### 3.3. System Configuration

Please refer to chapter 4.12 [sysconf] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'System Configuration' form on the right. The menu includes links for Network Interface, SIP Information, System Configuration (highlighted), PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password, ROM Configuration, Flash Clean, Commit Data, and Reboot System. The System Configuration form contains the following fields:

System Configuration	
Keypad DTMF Type:	<input checked="" type="radio"/> In-Band <input type="radio"/> RFC2833
RFC2833 Payload Type:	<input type="text" value="96"/>
Inter Digit Time:	<input type="text" value="3"/>
End of Dial Digit:	<input type="radio"/> NONE <input type="radio"/> * <input checked="" type="radio"/> #
<input type="button" value="OK"/>	

Keypad DTMF Type: set DTMF type. User can select DTMF type IP-Phone transmits.

RFC2833 Payload Type: change RFC2833 Payload type. This is for special request from the other site, if RFC2833 payload types of 2 sites are different, it may cause some problem of connection.

Inter Digit Time: Set the DTMF inter digit time (second)

To set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, IP-Phone will dial out all number pressed.

End of Dial Digit: select end of dialing key, e.g. set end of dial key as \* button, after finished pressing dialing number then press \* will dial out.

### 3.4. PPPoE Configure

Please refer to chapter 4.11[pppoe] command

<b>SIP-Phone Configuration Menu</b> <a href="#">Network Interface</a> <a href="#">SIP Information</a> <a href="#">System Configuration</a> <b><a href="#">PPPoE Configure</a></b> <a href="#">Voice Setting</a> <a href="#">Phone Book</a> <a href="#">DSCP Configuration</a> <a href="#">Password</a> <a href="#">ROM Configuration</a> <a href="#">Flash Clean</a> <a href="#">Commit Data</a> <a href="#">Reboot System</a>	PPPoE Device Information and Configuration	
	User Name:	pppoe
	Password:	*****
	Reboot After Remote Host Disconnection:	<input checked="" type="radio"/> On <input type="radio"/> Off
	Authenticate:	PAP
	Protocol:	TCP/IP
	Device:	PPP/PPPoE
	<input type="button" value="OK"/>	

User Name: Set PPPoE authentication User Name.

Password: Set PPPoE authentication password.

Reboot After Remote Host Disconnection: Enable/Disable auto reboot after PPPoE disconnection

If user enables this function, after PPPoE being disconnected, IP-Phone will automatically reboot to re-connect, and after reboot, if IP-Phone still can't get contact with server, IP-Phone will keep trying to connect. After re-connected, IP-Phone will also restart system. On the other hand, if user disables this function, IP-Phone won't reboot and keep trying to connect.

Other items: for reference only, cannot allow to be configured.

### 3.5. Voice Setting

Please refer to chapter 4.15 [voice] command

The screenshot displays the 'SIP-Phone Configuration Menu' with a sidebar on the left containing various configuration options. The 'Voice Setting' tab is selected, showing a table for codec priorities and several configuration fields.

Voice Setting					
Codec Priority	1st G.723.1	2nd G.729	3rd G.711mu-Law	4th G.711A-Law	5th G.729a
Frame Size	G.723.1 60ms	G.729a 60ms	G.729 60ms	G.711mu 40ms	G.711A 40ms
G.723 Silence Suppression:	<input type="radio"/> enable <input checked="" type="radio"/> disable				
Volume:	voice 32 input 32		DTMF 31		
Echo Cancelor:	<input checked="" type="radio"/> enable <input type="radio"/> disable				
Jitter Buffer:	Min. Delay 90		Max. Delay 150		
Optimized Factor (Jitter):	9				
OK					

Codec Priority: set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need. For example, user can only set first priority as G.723.1, and set the others as x, that means only G.723.1 is available.

Frame Size: User can set different packet size for each codec.

G.723 Silence Suppression: Enable / Disable sound compression and comfort noise generation. It is only for codec G.723.1

Volume: Adjust the volume in "Voice" (sending out); "Input" (receiving); "DTMF" (DTMF sending out).

Echo Cancelor: Enable / Disable (suggested always Enable this function).

Jitter Buffer: Set Min. Delay and Max. Delay of Jitter Buffer for voice packets.

Optimized Factor (Jitter): Set Optimized Factor of voice, this is for advanced user only, please contact with your distributor before making any change.

## 3.6. Phone Book

Please refer to chapter 4.10 [pbook] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left sidebar, with the 'Phone Book' option selected. The main content area is divided into two sections: 'Phone Book' and 'New Record'.

**Phone Book**

Index	Name	IP_Address	e164

**New Record**

Index	Name	IP Address	E164 No.
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

Below the 'New Record' form are two buttons: 'Add Data' and 'Delete Data'.

Add Data: User can specify 20 sets of phone book via web interface. Please input index, Name, IP Address and E.164 number of the destination device.

Delete Date: User can delete any configured phone book data by index.



### 3.7.DSCP Configure

Please refer to chapter 4.16 [tos] command

**SIP-Phone Configuration Menu**

- [Network Interface](#)
- [SIP Information](#)
- [System Configuration](#)
- [PPPoE Configure](#)
- [Voice Setting](#)
- [Phone Book](#)
- [DSCP Configuration](#)**
- [Password](#)
- [ROM Configuration](#)
- [Flash Clean](#)
- [Commit Data](#)
- [Reboot System](#)

**DiffServ Code Point(DSCP) Configuration**

=== Signal Packet ===

☐ Assured Forwarding (AF) PHB      Delay Priority :       Drop Precedence :

☐ Expedited Forwarding(EF) PHB

☒ Default

☐ User Assign Special DSCP Code:

=== RTP Packet ===

☐ Assured Forwarding (AF) PHB      Delay Priority :       Drop Precedence :

☐ Expedited Forwarding(EF) PHB

☒ Default

☐ User Assign Special DSCP Code:

Set Signal or RTP Packet DSCP value:

Assured Forwarding (AF) PHB: Select Delay priority and Drop Precedence

Expedited Forwarding (EF) PHB: Select TOS value as EF

Default: Select TOS value as 0

User Assign Special DSCP Code: User can set other unspecified value here.

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Code Point (DSCP) of the DS field in the IP packet header, and map each Code Point to a corresponding egress traffic priority. As per the definition in RFC2474, the DS field is Type-of-Service (TOS) octet in IPv4. The recommended DiffServ Code Point is defined in RFC2597 to classify the traffic into different service classes. The mapping of Code Point value of DS-field to egress traffic priorities is shown as follows.

DROP Precedence	Class #1	Class #2	Class #3	Class #4
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

Please refer to RFC standard documents for more information about what is DSCP.

## 3.8.Password

Please refer to chapter 4.19 [password] command

The screenshot displays a web-based configuration interface for a SIP-Phone. On the left is a vertical sidebar titled "SIP-Phone Configuration Menu" containing several menu items: "Network Interface", "SIP Information", "System Configuration", "PPPoE Configure", "Voice Setting", "Phone Book", "DSCP Configuration", "Password" (which is highlighted with a dashed border), "ROM Configuration", "Flash Clean", "Commit Data", and "Reboot System". The main content area on the right is titled "Password" in a blue header. It contains a form with a dropdown menu on the left showing "root". To the right of the dropdown are three input fields labeled "Current Password:", "New Password:", and "Confirm New Password:". At the bottom of the form are two buttons: "CHANGE" and "ABORT".

Password	
root	Current Password: <input type="text"/>
	New Password: <input type="text"/>
	Confirm New Password: <input type="text"/>
<input type="button" value="CHANGE"/> <input type="button" value="ABORT"/>	

Change: First select login name as root or administrator, then enter current password, new password and confirm new password again to set new password.

Abort: Press abort will clean all inputs.

### 3.9. Rom Configuration

Please refer to chapter 4.18 [rom] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'ROM Configuration' form on the right. The menu includes links for Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password, ROM Configuration (highlighted), Flash Clean, Commit Data, and Reboot System. The ROM Configuration form contains the following fields:

ROM Configuration	
FTP/TFTP server IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Target File name:	<input type="text"/>
Method:	<input type="text" value="TFTP"/>
FTP Login:	name <input type="text"/> passwd <input type="text"/>
Target File Type:	<input type="text" value="Application Image"/>
<input type="button" value="OK"/>	

FTP/TFTP Server IP Address: Set TFTP server IP address

Target File name: Set file name prepared to upgrade

Method: Select download method as TFTP or FTP

FTP Login: Set FTP login name and password

Target File Type: Select which sector of IP-Phone to upgrade

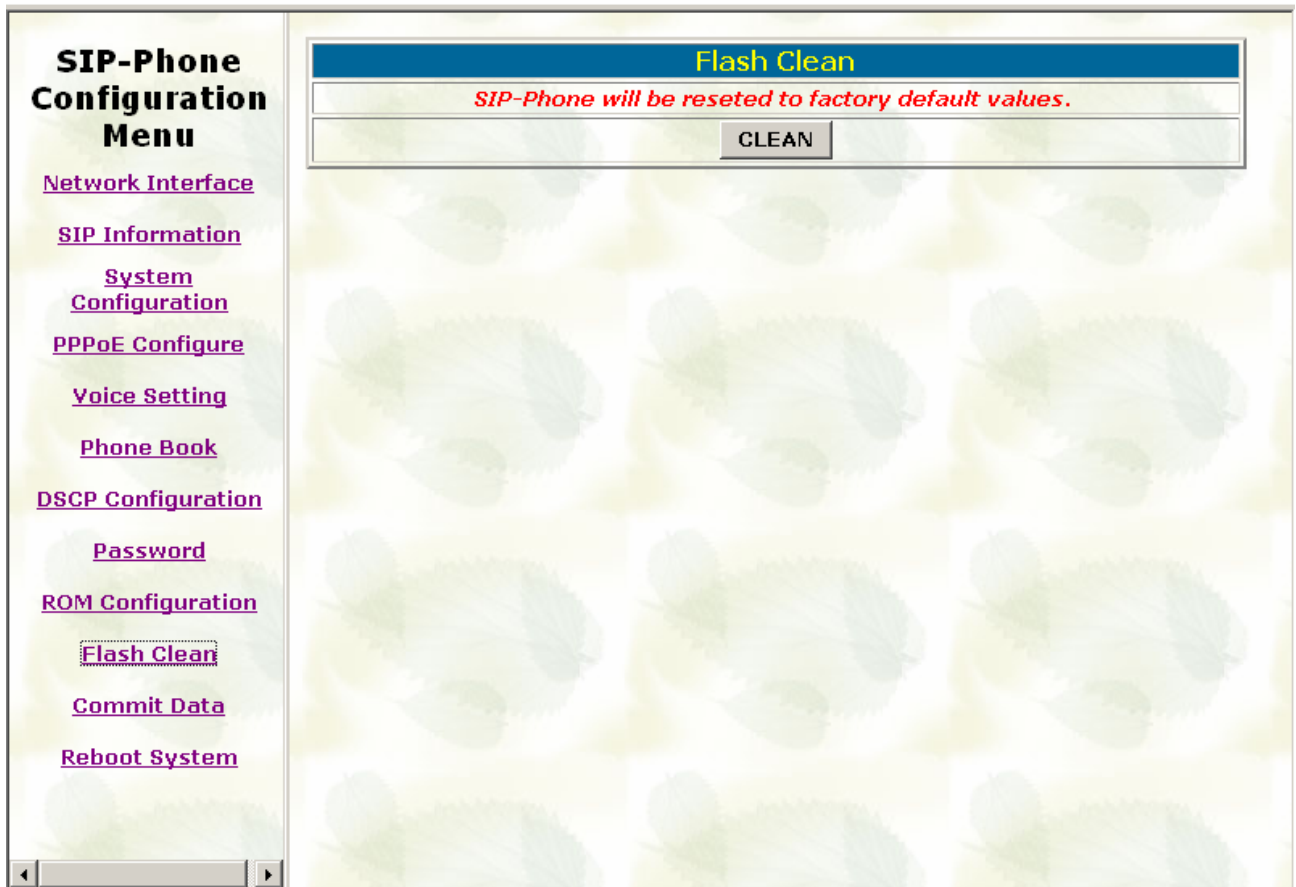
**Note:**

After 2mb file download is finished, all configurations might change to default values, user has to configure again.

After upgrade Application, please remember to execute Flash Clean, which will clean all configurations become factory values except IP address.

### 3.10. Flash Clean

Please refer to 4.5 [flash] command.



Press CLEAN will clean all configurations of IP-Phone and reset to factory default value.

Note: User must re-configure all commands all over again (except Network Configure) once execute this function,

### 3.11. Reboot System

The screenshot displays the 'SIP-Phone Configuration Menu' on the left sidebar, which includes links for Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password, ROM Configuration, Flash Clean, Commit Data, and Reboot System. The main content area shows the 'Reboot SIP-Phone' dialog box. This dialog has a blue header with the title 'Reboot SIP-Phone'. Below the header, it contains the text 'It will take 40 seconds to reboot. (remember to COMMIT data before reboot!)'. At the bottom of the dialog is a 'REBOOT' button.

Reboot SIP-Phone
It will take 40 seconds to reboot. (remember to COMMIT data before reboot!)
<input type="button" value="REBOOT"/>

Press reboot will reset IP-Phone.

Note: To execute reboot via web browser, IP-Phone will automatically save all data before reboot. To execute reboot via TELNET command, please remember to do Commit Data before Reboot System.

## 4. Configuring the IP-Phone Telnet command lines

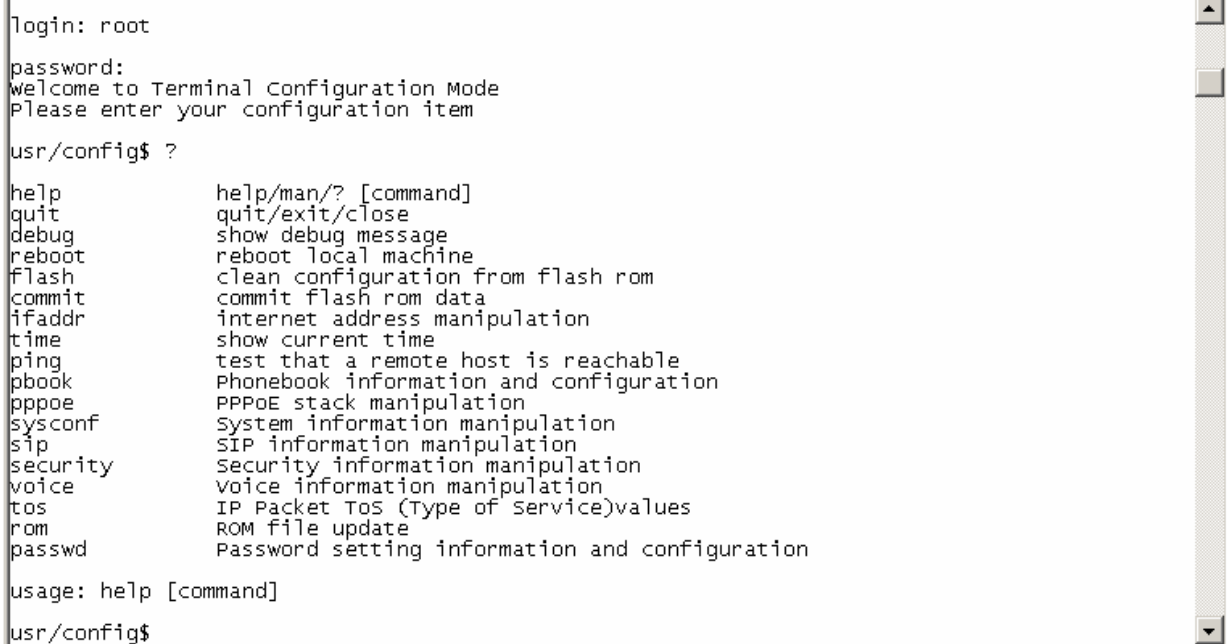
After setting the IP Address of IP-Phone and reboot, (please refer to LCD Menu: 5-3.4.5), user can enter into Telnet command lines.

### Note:

- After user enter IP-Phone configuration via telnet, please use login: "root", password: null, press enter to enter command lines.
- User must input lower-case command, but contents of configurations such as SIP alias or user name etc, user can set as capital case.
- After any change of configuration, please remember to do **commit** command to save changes and then **reboot** command to reboot system.

### 4.1. [help] command

Type **help** or **man** or **?** to display all the command lists. The following figure is shown all commands of IP-Phone .



```
Login: root
password:
Welcome to Terminal Configuration Mode
Please enter your configuration item
usr/config$ ?

help          help/man/? [command]
quit          quit/exit/close
debug         show debug message
reboot        reboot local machine
flash         clean configuration from flash rom
commit        commit flash rom data
ifaddr        internet address manipulation
time          show current time
ping          test that a remote host is reachable
pbbook        Phonebook information and configuration
pppoe         PPPoE stack manipulation
sysconf       System information manipulation
sip           SIP information manipulation
security       Security information manipulation
voice         voice information manipulation
tos           IP Packet ToS (Type of Service)values
rom           ROM file update
passwd        Password setting information and configuration

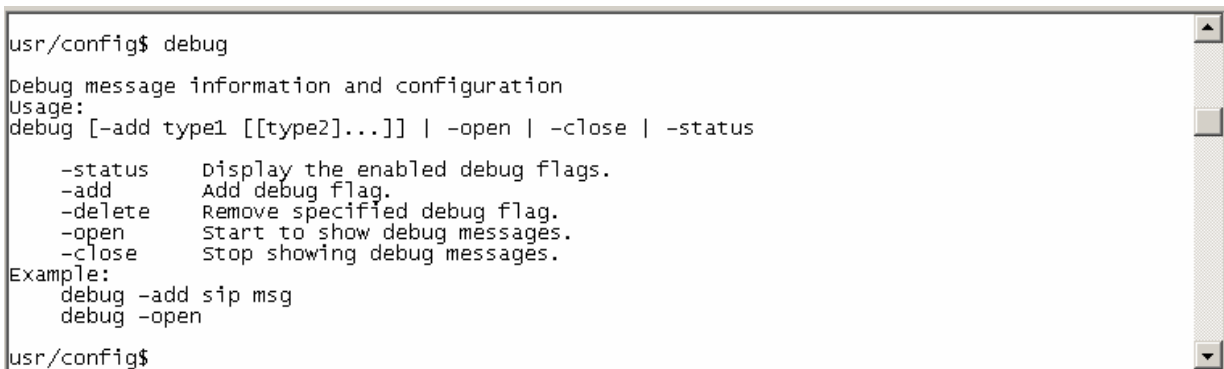
usage: help [command]
usr/config$
```

### 4.2. [quit] command

Type **quit** / **exit** / **close** will logout IP-Phone and close Telnet Program.

### 4.3. [debug] command

This command is for engineers to debug system of IP-Phone. User can add debug flag via command **debug –add “debug flags”**, and then start debug function via command **debug –open**. When IP-Phone is working on screen will display related debug messages. Most frequently used debug flag are “sip”, “fsm”, “msg”...etc.



```
usr/config$ debug
Debug message information and configuration
Usage:
debug [-add type1 [[type2]...]] | -open | -close | -status
    -status      Display the enabled debug flags.
    -add         Add debug flag.
    -delete      Remove specified debug flag.
    -open        Start to show debug messages.
    -close       Stop showing debug messages.
Example:
debug -add sip msg
debug -open
usr/config$
```

### 4.4. [reboot] command

After typing commit command, type reboot to restart the IP-Phone .

Sometimes after user type reboot, on terminal screen will display: "Data modified, commit to flash rom?" which means IP-Phone will record call history or not. (Ex. REDIAL, outgoing and incoming call data)

### 4.5. [flash] command

This command will clean the configuration stored in the flash rom to default value and reboot the IP-Phone .

#### Note:

- After user upgrade new software version, suggested to execute this command to make sure new software work well on IP-Phone
- To execute the command **flash –clean**, all configuration of IP-Phone stored in flash will be cleaned. It is authorized for the user whose login name is “root” only.



```
usr/config$ flash
Flash memory information and configuration
Usage:
flash -clean

Note:
  This command will clean the configuration stored in
  the flash and reboot it.

usr/config$
```

## 4.6. [commit] command

Save any changes after configuring the IP-Phone .

```
usr/config$ commit

This may take a few seconds, please wait....

Commit to flash memory ok!
usr/config$
```

## 4.7. [ifaddr] command

Configure and display the IP-Phone IP information.

- **-print**: print out all current configurations of ifaddr command.
- **-ip, -mask, -gate**: Set IP-Phone IP Address, subnet mask and default gateway respectively.
- **-ipmode**: Set IP-Phone network mode to be Fixed IP, DHCP or PPPoE.  
When User set IP mode to be fixed IP, please set IP, subnet Mask, default gateway as mentioned in item 2.  
If User set IP mode to be DHCP, IP-Phone will search for DHCP server to capture IP address after reboot.  
If user set IP mode to be PPPoE, please remember to set related parameters under **[pppoe]** command.
- **-sntp**: When SNTP server is available, enable IP-Phone SNTP function and assign SNTP server IP address so that IP-Phone can capture current time from SNTP server. (**ifaddr -sntp 1 "xxx.xxx.xxx.xxx"**)
- **-dns**: User can set Domain Name Server IP address. Once IP-Phone can connect with DNS server, user can specify URL address instead of IP address for Proxy Server and phone book IP address...etc.
- **-timezone**: User can set different time zone according to the location IP-Phone is. For example, in Taiwan the time zone should be set as 8, which means GMT+8. (GMT-8: **ifaddr -timezone -8**)

- **-ipsharing:** If IP-Phone is behind a IP-sharing , user must enable IP sharing function and specify public IP address.(ifaddr **-ipsharing 0 / 1** “**public IP address of IP sharing**” , 0 for disable and 1 for enable)

**Note:**

**Some Proxy servers support endpoint behind NAT function, in this case IP-Phone doesn't have to enable IP sharing function, please contact with your Proxy Server vendor.**

- **-upnp:** enable / disable UPnP function. If the IP sharing or NAT device supports UPnP, user can enable UPnP function so that IP-Phone will automatically connect with NAT device without configuration in IP Phone and NAT device.

```
usr/config$ ifaddr

LAN information and configuration
Usage:
ifaddr [-print][[-dhcp used]][-sntp mode [server]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-dns index [dns server address]]

    -print      Display LAN information and configuration.
    -ip         Specify ip address.
    -mask       Set Internet subnet mask.
    -gate       Specify default gateway ip address
    -ipmode     Set get IP mode(0:Fixed IP/1:DHCP/2:PPPoE)
    -sntp       Set SNTP server mode and specify IP address.
    -dns        specify IP address of DNS Server.
    -timezone   Set local timezone.
    -ipsharing  Specify usage of an IP sharing device and specify IP address.
    -upnp       specify the upnp mode of ipsharing(0:off/1:on)

Note:
Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).
Example:
ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
ifaddr -ipmode 1
ifaddr -sntp 1 210.59.163.254
ifaddr -ipsharing 1 210.59.163.254
ifaddr -upnp 1
ifaddr -dns 1 168.95.1.1

usr/config$
```

## 4.8. [time] command

When IP-Phone enable SNTP function and be able to connect with SNTP server, type **time** command will show the current time retrieved from SNTP server.

```
usr/config$ time

Current time is TUE FEB 03 14:50:08 2004

usr/config$
```

## 4.9. [ping] command

Command ping can test which the IP address is reachable or not.

### Usage: ping “IP address”

The message will display packets transmitting condition or no answer from the IP address.

```
usr/config$ ping
usr/config$ ping 192.168.2.107
PING 192.168.2.107: 56 data bytes
64 bytes from 192.168.2.107: icmp_seq=0. time=5. ms
64 bytes from 192.168.2.107: icmp_seq=1. time=0. ms
64 bytes from 192.168.2.107: icmp_seq=2. time=0. ms
64 bytes from 192.168.2.107: icmp_seq=3. time=0. ms
----192.168.2.107 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)  min/avg/max = 0/1/5
usr/config$ ping 192.168.99.99
PING 192.168.99.99: 56 data bytes
no answer from 192.168.99.99
usr/config$
```

## 4.10. [pbook] command

This command is functional both in Proxy mode and Peer-to-Peer mode. In proxy mode, use speed dial or 10 DL button will dial out e.164 number in phone book. In the other hand, in peer-to-peer mode, IP-Phone will dial out IP address.

- **-print:** display phone book data. User can print all data in phone book by command (**pbook -print**). Furthermore, user can also print only a section of data by indicate parameter “start index” and “end index” (**pbook -print “start index” “end index”**). If parameter “end index” is omitted, only record “start index” will be displayed. (**pbook -print “start prefix”**).
- **-add:** add a new record in phone book table by giving name, IP address, and e.164 number of callee endpoint.  
(**pbook -add name “X” ip “xxx.xxx.xxx.xxx” e164 “X”**)
- **-search:** search any record in the phone book according to IP address, name or e164 number
- **-delete:** delete a record of certain listed index in phone book table. (**pbook -delete “index number”**)
- **-insert:** insert an record in specified index of phone book.
- **-modify:** modify record of a certain index in phone book . Please notice that the name, IP address and e164 number must be modified together; user cannot just modify one parameter only.  
(**pbook -modify “index” name “X” ip “xxx.xxx.xxx.xxx” e164 “X”**)

```

usr/config$ pbook

Phonebook information and configuration
Usage:
pbook [-print [start_record] [end_record]]
pbook [-add [ip ipaddress] [name Alias] [e164 phonenum]]
pbook [-search [ip ipaddress] [name Alias] [e164 phonenum]]
pbook [-insert [index] [ip ipaddress] [name Alias] [e164 phonenum]]
pbook [-delete index]
pbook [-modify [index] [ip ipaddress] [name Alias] [e164 phonenum]]

    -print      Display phonebook data.
    -add        Add an record to phonebook.
    -search     Search an record in phonebook.
    -delete     Delete an record from phonebook.
    -insert     Insert an record to phonebook in specified position.
    -modify     Modify an exist record.

Note:
If parameter 'end_record' is omitted, only record 'start_record' will be display.
If both parameters 'end_record' and 'start_record' are omitted, all records will be display.
Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
Range of index setting value (1~100),
Example:
pbook -print 1 10
pbook -print 1
pbook -print
pbook -add name Test ip 210.59.163.202 e164 1001
pbook -insert 3 name Test ip 210.59.163.202 e164 1001
pbook -delete 3
pbook -search ip 192.168.4.99
pbook -modify 3 name Test ip 210.59.163.202 e164 1001

usr/config$

```

## 4.11. [pppoe] command

- **-print**: display all current configurations and information.
- **-id**: to set PPPoE authentication user name.
- **-pwd**: to set PPPoE authentication password.
- **-reboot**: Choose enable or disable this function. If user enables this function, after PPPoE disconnected, IP-Phone will automatically reboot to re-connect, and after reboot, if IP-Phone still can't connect with server, IP-Phone will keep trying to connect. On the other hand, if user disables this function, IP-Phone won't reboot and keep trying to connect. (**pppoe -reboot 0 / 1**)

```

usr/config$ pppoe

PPPoE device information and configuration
Usage:
pppoe [-print]
pppoe [-id username][-pwd password]

    -print      Display PPPoE device information.
    -id         Connection user name.
    -pwd        Connection password.
    -reboot     Reboot after remote host disconnection.

usr/config$

```

## 4.12. [sysconf] command

- **-print**: display all current configurations.
- **-idtime**: set the duration(in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, IP-Phone will dial out all number pressed.
- **-keypad**: set DTMF type .User can select DTMF type IP-Phone receive and transmit.(sysconf –keypad 0/1 , 0 for in band ,1 for RFC2833.)
- **-2833type**: change RFC2833 Payload type.
- **-eod**: select end of dialing key, e.g. set end of dial key as “\*” button , after finished pressing dialing number then press “\*” will dial out. (sysconf –eod 0/1/2 , 0 for no end of dial key , 1 for “\*” button, 2 for “#” button )
- **–service**: set IP Phone to be normal mode or under hotline mode. (sysconf –service 0/1, 0 for normal service, 1 for hotline service.)

### Note:

**To set IP-Phone as hotline mode, user has to set IP-Phone as Peer-to-Peer mode, and hotline table under bureau command.**

```
usr/config$ sysconf
System information and configuration
Usage:
sysconf [-idtime digit] [-keypad dtmf]
        [-2833type type] [-eod digit]
sysconf -print
-print          Display system overall information and configuration.
-idtime         Inter-Digits time.(1~10 sec)
-keypad         Select DTMF type: 0=In-band,
                1=RFC2833.
-2833type       RFC2833 Payload Type (range:96~128 inter-used:100,102~105)
-eod            End of Dial Digit setting(0: NONE, 1: *, 2: #)
Example:
sysconf -keypad 0 -eod 2
usr/config$
```

## 4.13. [sip] command

- **-print**: display all current configurations.
- **-mode**: configure IP-Phone as Proxy or Peer-to-Peer Mode.  
Usage: sip –mode 0/1(1 for Proxy mode, 0 for Peer-to-Peer mode)
- **-px**: set proxy server IP address or URL address (sip –px “IP address or URL of Proxy server”).
- **-pxport**: set listening port of Proxy server.
- **-outpx**: set IP address of outbound proxy server. After user set outbound proxy, all

packets from IP-Phone will be sent to outbound proxy server.

- **-prefix:** set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, IP-Phone will send out messages as Account Name @"IP address of Proxy", if user set prefix as abc, IP-Phone will send out as abc123@"IP address of Proxy". This function is for special proxy server.
- **-line:** identify one number for the IP-Phone to register to the Proxy (SIP -line "line number").

**Note: In proxy mode please remember to set user account information under security command.**

- **-pbsearch:** enable/disable phone book search function under Proxy Mode. If user enabled this function, IP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, IP-Phone will make call to related IP address.
- **-expire:** set expire time of registration. IP-Phone will keep re-registering to proxy server before expire timed out.
- **-port:** set listening UDP port of IP-Phone.
- **-rtp:** set RTP port number. IP-Phone will use this port to send and receive voice.

```
usr/config$ sip
SIP stack information and configuration
Usage:
sip [-mode pxmode]
sip [-px address] [-prefix prefixstring]
    [-pxport ProxyPort] [-outpx address] [-line number]
    [-expire t1] [-port udpPort] [-rtp rtpPort]
sip -print

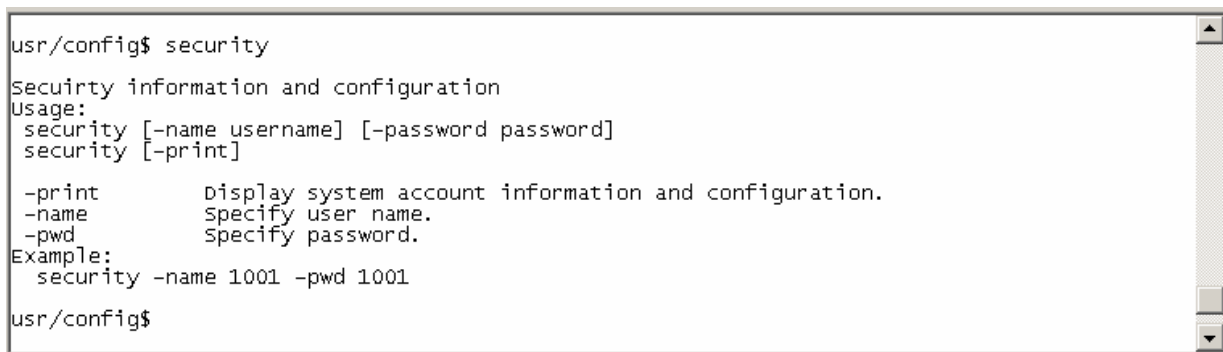
    -print      Display SIP stack information and configuration.
    -mode       Configure as Proxy mode or Peer-to-Peer mode.
    -px         Proxy server address. (Proxy IPv4 address or Proxy dns name)
    -pxport     Proxy server port. (the port of proxy)
    -outpx      OutBound Proxy server address. (Proxy IPv4 address or Proxy dns name)
                specify as null)
    -prefix     Specify prefix string, use it when UserID contains alphabets
                (if UserID uses numerals, specify as null)
    -line       TEL Phone number.
    -pbsearch   Search phone book      0:off/1:on.
    -expire     The relative time after which the message expires(0 ~ (2^31-1))
    -port       SIP local UDP port number (5060~5070), Default: 5060
    -rtp        RTP receive port number (2326~65534), Default: 16384
Example:
    sip -mode 1
    sip -px 210.59.163.171 -line 70
usr/config$
```

## 4.14. [security] command

- **-print:** display all current configurations.
- **-name:** set user ID of IP-Phone for registering. User can set user name and

password for registering. If password is no need, please set user name the same as line number or IP-Phone won't register successfully.

- **-pwd**: set account password for registering.



```
usr/config$ security
Security information and configuration
Usage:
security [-name username] [-password password]
security [-print]

-print          Display system account information and configuration.
-name          Specify user name.
-pwd           Specify password.
Example:
security -name 1001 -pwd 1001
usr/config$
```

## 4.15. [voice] command

The voice command is associated with the voice codec setting information.

- **-print**: display voice codec information and configuration.
- **-send**: three voice packet size can be configured as 20 ms, 40 ms or 60 ms.(only 30 and 60 ms for G.723.1)
- **-priority**: set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need, for example, voice -priority g723 or voice -priority g723 711a g711u g729 g729a means IP-Phone can support only one codec up to five codecs.
- **-volume**: There are three types can be adjustable, voice volume, input gain and DTMF volume. Voice volume means the volume user can hear, input gain means the volume the other side can hear from IP-Phone, DTMF means DTMF transmitting volume. (voice -volume voice "value of volume", voice -volume input "value of volume", voice -volume dtmf "value of volume")

### Note:

If value of volume set as 0 means -32db, 1 means -31db...etc.

- **-nscng**: enable or disable sound compression and comfort noise generation. It is only for codec G.723.1. (0 for off, 1 for on)
- **-echo**: enable or disable echo cancellation function .
- **-mindelay**: set minimum delay of jitter buffer(0~150)
- **-maxdealy**: set maximum delay of jitter buffer(0~150)

**Note: It is for advanced administrator use only. Please ask your**

**distributor before changing any settings of this command.**

```
usr/config$ voice
Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G729 ms] [G711U ms] [G711A ms] [G729A ms] ]
      [-volume line [voice level] [input level] [dtmf level]]
      [-nscng [G711U used1] [G711A used2] [G723 used3]]
      [-echo used] [-mindelay t1] [-maxdelay t2] [-optfactor f]
voice -print
voice -priority [G723] [G729] [G711U] [G711A] [G729A]

    -print      Display voice codec information and configuration.
    -send       Specify sending packet size.
                G.723 (30/60/90 ms)
                G.729 (20/40/60 ms)
                G.711U (20/40/60 ms)
                G.711A (20/40/60 ms)
                G.729A (20/40/60 ms)
    -priority   Priority preference of installed codecs.
                G.723
                G.729
                G.711U
                G.711A
                G.729A
    -volume     Specify the following levels:
                voice volume (0~63, default: 30),
                input gain (0~63, default: 30),
                dtmf volume (0~31, default: 23),
    -nscng      No sound compression and CNG. (G.723.1 only, on=1, off=0).
    -echo       Setting of echo canceller. (On=1, Off=0, per port basis).
    -mindelay   Setting of jitter buffer min delay. (0~150, default: 90).
    -maxdelay   Setting of jitter buffer max delay. (0~150, default: 150).
Example:
voice -send g723 60 g729 60 g711u 60 g711a 60 g729a 60
voice -volume 1 voice 20 input 32 dtmf 27
voice -echo 1
usr/config$
```

## 4.16. [tos] command

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Code Point (DSCP) of the DS field in the IP packet header, and map each Code Point to a corresponding egress traffic priority. As per the definition in RFC2474, the DS field is Type-of-Service (TOS) octet in IPv4. The recommended DiffServ Code Point is defined in RFC2597 to classify the traffic into different service classes. The mapping of Code Point value of DS-field to egress traffic priorities is shown as follows.

- High priority with DS-field.
 

Expected Forwarding (EF)	101110	====>	46 (Decimal System)
Assured Forwarding (AF)	001010	====>	10 (Decimal System)
	010010	====>	18 (Decimal System)
	011010	====>	26 (Decimal System)
	100010	====>	34 (Decimal System)
- Low Priority with DS-field:
 

Assured Forwarding (AF)	001100	====>	12 (Decimal System)
	010100	====>	20 (Decimal System)



011100	====>	28 (Decimal System)
100100	====>	36 (Decimal System)
001110	====>	14 (Decimal System)
010110	====>	22 (Decimal System)
011110	====>	30 (Decimal System)
100110	====>	38 (Decimal System)
000000	====>	0 (Decimal System)

DROP Precedence	Class #1	Class #2	Class #3	Class #4
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

- **-print:** display all current configurations.
- **-rtptype:** set DSCP value of signaling packets from 0 to 63
- **-siptype:** set DSCP value of RTP packets from 0 to 63

**Note:**

- This command won't be functional until network environment can be capable with DSCP function.
- **tos -rtptype 14 -sigtype 10** is top priority of package.

```
usr/config$ tos
IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
tos -print
tos [-rtpreliab mode]
tos -print
Example:
tos -rtptype 10 -sigtype 0
usr/config$ █
```

## 4.17. [bureau] command

- **-print**: show all current configurations.
- **-hotline**: set hotline table. User can set hotline function to specify one IP address for IP-Phone to dial out directly. Once user picks up IP-Phone, it will automatically dial out to the assigned IP address and number.  
(bureau -hotline "IP address" "phone number of remote site")

### Note:

To set IP-Phone as hotline mode, user has to set IP-Phone as **Peer-to-Peer** mode, use **sysconf -service 2** command.

```
usr/config$ bureau
Bureau line setting information and configuration
Usage:
bureau [-hotline [Port DestIP TELnum]]
bureau -print
    -print    Display Bureau line information and configuration.
    -hotline  Set Hot line information. (Port range: 1~6)
Note:
Hotline feature should be used together with:
    $sysconf -service 2 (HotLine service)
    $sip -mode 1 (peer-to-peer mode)
Example:
    bureau -hotline 1 192.168.4.69 628
usr/config$
```

## 4.18. [rom] command

- **-print**: show all current configurations and version information.
- **-app,-boot, -dsptest, -dspcore, -dspapp,:** upgrade main boot code, main application code, DSP testing code, DSP kernel code, DSP application code, Ring Back Tone PCM file and Hold Tone .

### Note:

**After upgrade Application, please remember to execute flash -clean command, which will clean all configurations become factory values except IP address.**

- **-boot2m**: to upgrade 2mb rom file, which includes all firmware file mentioned in item 2.
- **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading firmware rom file.

- **-f**: the file name prepared for upgrading is necessary as well.
- **-method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
- **-ftp**: specify user name and password for FTP download method.

For example: User prepares to upgrade the latest app rom file – sip1p.100, the TFTP server is 192.168.1.1.

rom -app -s 192.168.1.1 -f sip1p.100

```
usr/config$ rom
ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp]
-s TFTP/FTP server ip -f filename
rom -print
rom -print      show versions of rom files. (optional)
rom -app        update main application code(optional)
rom -boot       update main boot code(optional)
rom -boot2m     update 2M code(optional)
rom -dsptest    update DSP testing code(optional)
rom -dspcore    update DSP kernel code(optional)
rom -dspapp     update DSP application code(optional)
rom -s          IP address of TFTP/FTP server (mandatory)
rom -f          file name(mandatory)
rom -method     download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
rom -ftp        specify username and password for FTP
Note:
This command can run select one option in 'app', 'boot',
, 'dsptest', 'dspcore', and 'dspapp'.
Example:
rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin
usr/config$
```

Command rom -print can show current version installed in IP-Phone .

```
usr/config$ rom -print
Download Method : TFTP
Boot Rom       : lp201_boot.101
Application Rom : lp201_sip0202.BIN
  DSP App      : 48302ce3.140
  DSP Kernel   : 48302ck.140
  DSP Test Code : 483cbit.bin
usr/config$
```

## 4.19. [passwd] command

For security protection, user has to input the password before entering application user/config mode. Two configurations of login name/password are supported by the system.

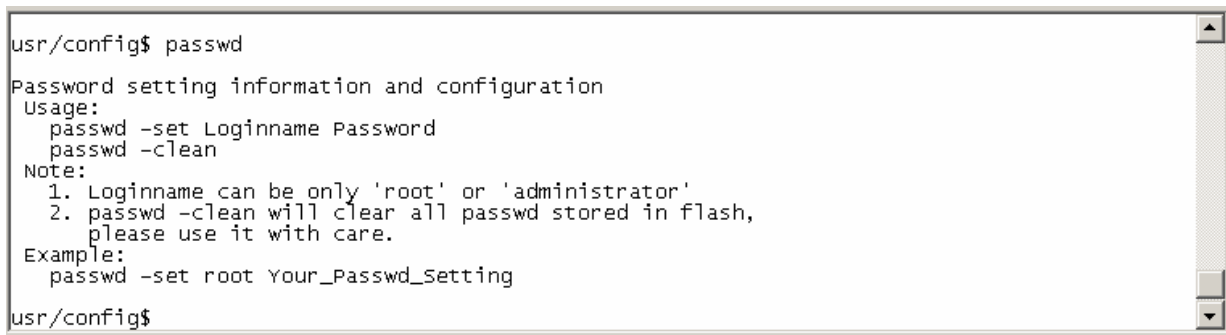
- **-set**: set password of “root” users or “administrator” users. (passwd -set root/administrator “password”)
- **-clean**: clean up password restored before, and user can login :”root/administrator”,

password: "press enter".

User who requests authorization to execute all configuration commands needs to login with "root". If a user login with "administrator", two commands are not functional:

password -set root: set password of login : "root".

passwd -clean: clean up password restored before, and user can login : "root/administrator",  
password: "press enter".



```
usr/config$ passwd
Password setting information and configuration
Usage:
  passwd -set Loginname Password
  passwd -clean
Note:
  1. Loginname can be only 'root' or 'administrator'
  2. passwd -clean will clear all passwd stored in flash,
     please use it with care.
Example:
  passwd -set root Your_Passwd_Setting
usr/config$
```

## 5. Upgrading Software on the IP-Phone

IP-Phone supports three methods to upgrade the new version. All methods are necessary to prepare the TFTP or FTP program on the host PC as TFTP/FTP server. After installing TFTP/FTP program on one PC and connecting to network, IP-Phone is ready to be upgraded.

### 5.1.From LCD Phone Menu

Please choose the 6→2 selection-Firmware Upgrade. Press OK to enter into the sub-selection as below.

Firmware Update

#### 1. Download method

There are two methods to download new version file, please move the ">" symbol by press ← or → on the keypad to select TFTP or FTP method, then press **OK** to confirm it.

#### 2. FTP/TFTP Sever

User has to offer one TFTP/FTP server IP Address and set this IP Address via keypad. The IP Address is necessary for upgrading IP-Phone new application rom file.

#### 3. Set FTP user account (\*\*)

User has to input user name and password for FTP server login .It is necessary for upgrading IP-Phone new application rom file via FTP method.

#### 4. File Name

User has to press the file name of new application rom file prepared for upgrading

#### 5. Start to Upgrade

Select YES or NO to start upgrade. After download is finished, press OK then IP-Phone will ask if need to reboot.

#### 6. Firmware Version

Show versions of all software and hardware. (\*\*)

**Note:**

**Download via LCD command can only upgrade new application rom file.**

## 5.2. Web Management

Please refer to chapter 4.18 [rom] command

The screenshot displays the 'SIP-Phone Configuration Menu' on the left and the 'ROM Configuration' form on the right. The menu includes links for Network Interface, SIP Information, System Configuration, PPPoE Configure, Voice Setting, Phone Book, DSCP Configuration, Password, ROM Configuration (highlighted), Flash Clean, Commit Data, and Reboot System. The ROM Configuration form contains the following fields:

ROM Configuration	
FTP/TFTP server IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Target File name:	<input type="text"/>
Method:	<input type="text" value="TFTP"/>
FTP Login:	name <input type="text"/> passwd <input type="text"/>
Target File Type:	<input type="text" value="Application Image"/>
<input type="button" value="OK"/>	

7. FTP/TFTP Server IP Address: Set TFTP server IP address
8. Target File name: Set file name prepared to upgrade
9. Method: Select download method as TFTP or FTP
10. FTP Login: Set FTP login name and password
11. Target File Type: Select which sector of IP-Phone to upgrade

**Note:**

After 2mb file download is finished, all configurations might change to default values, user has to configure again.

After upgrade Application, please remember to execute Flash Clean, which will clean all configurations become factory values except IP address.

## 5.3.Remote Control: Telnet

Please refer to chapter 4.18 [rom] command

1. **-print**: show all current configurations and version information.
2. **-app,-boot, -dsptest, -dspcore, -dspapp**: upgrade main boot code, main application code, DSP testing code, DSP kernel code, DSP application code, Ring Back Tone PCM file and Hold Tone .

### Note:

After upgrade Application, please remember to execute flash –clean command, which will clean all configurations become factory values except IP address.

3. **-boot2m**: to upgrade 2mb rom file, which includes all firmware file mentioned in item 2.
4. **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading firmware rom file.
5. **-f**: the file name prepared for upgrading is necessary as well.
6. **-method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
7. **-ftp**: specify user name and password for FTP download method.

For example: User prepares to upgrade the latest app rom file – sip1p.100, the TFTP server is 192.168.1.1.

**rom –app –s 192.168.1.1 –f sip1p.100**

```
usr/config$ rom
ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp]
-s TFTP/FTP server ip -f filename
rom -print
    -print      show versions of rom files. (optional)
    -app       update main application code(optional)
    -boot      update main boot code(optional)
    -boot2m    update 2M code(optional)
    -dsptest   update DSP testing code(optional)
    -dspcore   update DSP kernel code(optional)
    -dspapp    update DSP application code(optional)
    -s         IP address of TFTP/FTP server (mandatory)
    -f         file name(mandatory)
    -method    download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
    -ftp       specify username and password for FTP
Note:
This command can run select one option in 'app', 'boot',
, 'dsptest', 'dspcore', and 'dspapp'.
Example:
rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin
usr/config$
```

Command rom –print can show current version installed in IP-Phone .

```
usr/config$ rom -print
```

```
Download Method : TFTP  
  Boot Rom      : lp201_boot.101  
Application Rom : lp201_sip0202.BIN  
  DSP App       : 48302ce3.140  
  DSP Kernel    : 48302ck.140  
  DSP Test Code : 483cbit.bin  
usr/config$
```